REMARKS / ARGUMENTS

This letter is responsive to the Office Action mailed on October 19, 2005. Accordingly, the response is accompanied by a request for a three-month extension of time along with the required fees.

CLAIM AMENDMENTS

By this response, the Applicant has amended claims 1, 2, 13, 14, 18, 33, 42, 43 and 57. Claims 41 and 44 have been cancelled without prejudice. There are now 55 claims pending for this application.

The Applicant has amended claim 1 to more clearly recite the invention. Claim 1 now recites that the method comprises receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically controlling the configuration of said input/output characteristic for loudness normalization. Support for this claim amendment is in the passages at lines 14-29 on page 13, and lines 4-12 on page 19, as well as Figures 7A and 7B of the application as originally filed.

The Applicant has also amended claim 13 to more clearly recite the invention. Claim 13 now recites that the method includes receiving N user adjustable digital loudness normalization control signals from a user during operation for dynamically controlling the configurable composite input/output characteristic for loudness normalization, each of the loudness control signals corresponding to one of the frequency domain input signals. Support for this claim amendment is in the passages at lines 14-29 on page 13, lines 8-29 on page 15, and lines 4-12 on page 19, as well as Figure 7B of the application as originally filed.

The Applicant has amended claim 18 to correct an inadvertent error. On line 2 of claim 18, the text "is varied" has been corrected to read "are varied".

The Applicant has also amended claim 33 to more clearly recite the invention. Claim 33 now recites that the loudness normalization adjustment stage includes a signal controlling device that is manipulated by a user during operation to provide a user adjustable digital loudness normalization control signal for dynamic loudness normalization. Support for this claim amendment is in the passages at lines 14-29 on page 13, lines 8-29 on page 15, and lines 4-12 on page 19, as well as claims 41 and 44, and Figures 7A and 7B of the application as originally filed.

The Applicant has amended claims 2, 14 and 57 due to the amendments made to claims 1, 13 and 33. The Applicant has amended claims 2, 14 and 57 to properly refer back to the antecedent "a user".

The Applicant has amended the claim dependencies of claims 42 and 43 to depend from claim 33 instead of claim 41 since claim 41 was cancelled and the feature recited by claim 41 has been included in claim 33.

CLAIM OBJECTIONS

In the Office Action, the Examiner objected to claim 18 and stated that the phrase "is varied" should read "are varied".

In response, the Applicant has amended claim 18 according to the Examiner's suggestions.

CLAIM REJECTIONS – 35 USC § 102

In the Office Action, the Examiner rejected claims 1-41 and 44-57 under 35 USC 102(e) as being anticipated by Ishige (US 5,892,836).

With respect to claims 1, 13 and 33, the Examiner argued that Ishige discloses a

method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic (the Examiner referred to the abstract in Ishige) by: (a) converting the acoustic input signal into a digital acoustic input signal (the Examiner referred to element 12 in Fig. 7 of Ishige); (b) transforming the digital acoustic input signal into one or more frequency domain input signals (the Examiner referred to Fig. 6 and col. 7 lines 37-57 of Ishige); (c) detecting the magnitude of each of the one or more frequency domain input signals (the Examiner referred to element 21 in Fig. 7 of Ishige); (d) receiving a user adjustable digital loudness normalization control signal for dynamically controlling the configuration of said input/output characteristic (the Examiner referred to a memory in Ishige which previously stores hearing characteristics of a person to be fitted with the hearing aid, and can be programmed as a removable ROM or by communicating with a fitting device - see col. 6 lines 52-61 in Ishige); (e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal (the Examiner referred to the hearing compensating filter coefficient circuit in Ishige - see col. 7 lines 57-67 and col. 8 lines 1-10 in Ishige); (f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value (the Examiner referred to the digital audio signal supplied by the hearing compensating circuit – element 22 in Fig. 7 of Ishige); (g) transforming the one or more frequency domain output signals into a digital acoustic output signal (the Examiner referred to the output of element 22 in Fig. 7 of Ishige); and (h) converting the digital acoustic output signal into the analog acoustic output signal (the Examiner referred to element 13 in Fig. 7).

In response, the Applicant submits that claims 1, 13 and 33 of the subject application recite a user adjustable digital loudness normalization control signal that is received directly from a user during operation to dynamically control the configuration of an input/output characteristic for loudness normalization. This feature allows a user to dynamically alter the configuration characteristic for generating an acoustic output signal from an acoustic input signal during operation. Accordingly, the user can provide

the signal anytime during operation.

For example, for a hearing aid, the claimed invention allows a user to dynamically alter the loudness characteristics of the hearing aid, during operation, to optimize the output of the hearing aid from the user's perspective. This is done by the user directly actuating a mechanism to provide a signal during operation through various input means. Accordingly, as explained in the passage at lines 4-12 on page 19 of the application as filed, the claimed invention permits the use of minimal measurements during an initial fitting to first measure the user's auditory characteristics, and then allows the user to adjust the input/output characteristic of the loudness response of their hearing aid according to the user's preferences. This allows for the elimination of the time-consuming, and laborious task of measuring loudness data for the user, which is typically an inaccurate measurement.

Furthermore, loudness judgment is highly subjective and in some cases loudness ratings can change from day to day (for a variety of reasons). The user control allows the user to provide input to compensate for changes in the user's loudness perception over time.

Further, the Applicant emphasizes that the term during operation means that the user can adjust the input/output configuration anytime they wish by providing a signal through an input means. The user will typically adjust the input/output configuration after the initial fitting, but may also perform this operation during the initial fitting, under the guidance of a hearing aid professional.

In contrast, the Ishige reference is directed towards a hearing aid that provides a hearing compensating circuit that matches the hearing characteristics of the person to which the hearing aid was fitted with no input means that can be actuated by the user. The Applicant submits that the memory 24 referenced by the Examiner is used to store the static hearing characteristics of the user. While this information is used to determine the input/output response of the hearing compensating filter (see lines 52-61 in Col. 6 of

Ishige) during operation, the user cannot provide an input to dynamically, and in real-time, alter the performance of the hearing aid during operation because there is no such input means that a user can manipulate. Accordingly, once the Ishige hearing aid is programmed, the performance of the device cannot be dynamically adjusted during operation by the user.

The Applicant respectfully requests that the Examiner find a section in the Ishige reference, which teaches a user input that the user can use during operation to adjust the performance of the hearing aid, or withdraw this rejection.

Further, even if the hearing characteristics of the user are considered to be an input, as the Examiner is contending, this information is only provided at the time of fitting the hearing aid. Since this information is stored in the memory 24, it is clear that the user does not adjust the hearing aid to provide this information to the hearing aid. Rather, a hearing aid technician most certainly stores this information in the memory 24. Accordingly, it is clear that the user does not provide an input control signal to the Ishige hearing aid.

In addition, the Applicant reminds the Examiner that measuring loudness data for a particular hearing-impaired user is a time-consuming, laborious and often inaccurate step. Accordingly, a user most likely will want to adjust the input/output characteristics of the hearing aid during use. The user will be able to do so with a hearing aid based on the claimed subject invention. In contrast, a user will not be able to do so based on Ishige's hearing aid since there is no way that a user can provide any input control signal to the Ishige hearing aid after the initial fitting.

Based on the above discussion, it is clear that the Ishige reference does not teach each claimed feature in claims 1, 13 and 33 of the subject invention. Accordingly, the Applicant submits claims 1, 13 and 33 are not anticipated by the Ishige reference and the Applicant respectfully requests that the Examiner remove the 102(e) rejection of these claims.

In the Office Action, with respect to claims 2, 14, 34-37, 44 and 57, the Examiner argued that, in addition to the elements stated above regarding claims 1, 13, 33 and 41, that Ishige further discloses: adjusting the configurable input/output characteristic for at least one frequency band corresponding to the one or more frequency domain input signals by one of: increasing the level of said configurable input/output characteristic by a larger amount for lower level sounds compared to higher level sounds when a user adjusts the user adjustable digital loudness normalization control signal to increase the level of the analog acoustic output signal, and decreasing the level of said configurable input/output characteristic by a small amount for lower level sounds compared to higher level sounds when the user adjusts the user adjustable digital loudness normalization control signal to decrease the level of analog acoustic output signal. Specifically, the Examiner argued that the memory is programmed/fitted to the hearing characteristic of the user, and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid (Col. 7, lines 4-7 in Ishige). The Examiner further argued that it is inherent that the sounds the user perceives as being lower will be amplified greater than the sounds that the user perceives as being higher.

In response, the Applicant submits that while Ishige mentions using compression of the audio signal to match the narrowed dynamic range of the person fitted with the hearing aid in lines 4-7, col. 7, Ishige is otherwise silent on how this is done. The only thing that Ishige teaches on this subject, repeatedly, is that a hearing compensating filter coefficient setting circuit receives a frequency analysis result from an analyzer 21, and the hearing characteristics of the person to be fitted with the hearing aid from the memory 24 and coefficients for a plurality of imaginary linear phase FIR filters from the channel filter coefficient setting circuit. On the basis of the analysis result and the hearing characteristics of the person to be fitted with the hearing aid, the hearing compensating filter coefficient setting circuit determines a weight for each of the linear phase FIR filters and then weights the coefficients of the linear phase FIR filter by

multiplying the coefficients with the corresponding obtained weight (see line 58, col. 7 to line 21, col. 8).

The Applicant respectfully submits that the above characteristics of the Ishige hearing aid are clearly different than the features recited in claims 2, 14 and 57 of the subject invention. Furthermore these claims, as well as claims 34-37 each recite the effect of a user digital loudness normalization control signal, provided by a user, which, as previously shown, is not taught at all by Ishige. Accordingly, since Ishige does not teach that a user provides a digital loudness normalization control signal during operation, it is very clear that Ishige cannot possibly teach increasing the level of a configurable input/output characteristic by a larger amount for lower level sounds compared to higher level sounds <u>when a user adjusts the user adjustable digital loudness normalization control signal</u> to increase the level of the analog acoustic output signal, and decreasing the level of said configurable input/output characteristic by a small amount for lower level sounds compared to higher level sounds <u>when the user adjusts the user adjustable digital loudness normalization control signal</u> to decrease the level of analog acoustic output signal.

Regarding claims 4, 39, and 53, the Examiner argued that Ishige teaches calculating a corresponding gain value for one or more frequency domain input signals by means of a fitting formula wherein a parameter of the fitting formula is provided by the user adjustable digital loudness normalization control signal. The Examiner further argued that Ishige teaches that the hearing compensating filter coefficient setting circuit defines the coefficients for the filters in the hearing compensating circuit based on data that is supplied by the user through the fitting of the hearing aid (Fig. 7 and associated text in Ishige).

In response, the Applicant respectfully submits that claims 4, 39 and 53 refer to the user adjustable digital loudness normalization control signal, which can be provided by the user, via a suitable input means, during the operation of the hearing aid to dynamically modify the operation of the hearing aid. A user cannot provide such a signal with the

Ishige hearing aid. This is also supported by the Examiner's observation that the user data is supplied during fitting, not during operation, which occurs after fitting, which is the case in the subject claimed invention. Furthermore, Ishige does not teach fitting formulas but only coefficient tables or filter parameter tables (see Figs. 7 and 8 in Ishige).

Regarding claims 5, 22, 38 and 51, the Examiner argued that Ishige teaches calculating a corresponding gain value for one or more frequency domain input signals by means of a lookup table wherein information in the look-up table is retrieved based on the user adjustable digital loudness normalization control signal. The Examiner further argued that Ishige teaches that the hearing compensating filter refers to coefficients stored in the coefficient table and the hearing compensating circuit is configured to cause the input audio signal to match the narrowed dynamic range of the person fitted with the hearing aid.

In response, the Applicant submits that Ishige does not teach retrieving information from the filter parameter table 28 or the coefficient table 27 based on user input since the user cannot provide an adjustable digital loudness normalization control signal in the Ishige hearing aid. Furthermore, the Ishige hearing aid does not even use the user's hearing characteristics in selecting information from the tables 27 and 28 since Figures 7 and 8 of Ishige clearly show that the memory 24 does not provide any input to the channel filter coefficient setting circuit 25 that selects the information from the tables 27 and 28.

With regards to claims 7, 25, 40 and 54, the Examiner argued that Ishige teaches calculating a corresponding gain value for one or more frequency domain input signals by means of a fitting formula and a lookup table wherein information in the look-up table is retrieved based on the user adjustable digital loudness normalization control signal. The Examiner further argued that Ishige teaches that the hearing compensating filter refers to coefficients stored in the coefficient table and the hearing compensating circuit

is configured to cause the input audio signal to match the narrowed dynamic range of the person fitted with the hearing aid.

Based on the explanations above, the Applicant respectfully submits that Ishige does not teach the features recited in claims 7, 25, 40 and 54. Ishige does not teach a user adjustable digital loudness normalization control signal, or even an input means that the user can use during operation to provide input to the hearing aid. Furthermore, the connections of the memory unit to the other components in the Ishige hearing aid shown in Figures 7 and 8 clearly show that no user information is used in selecting information from the tables 27 and 28.

With regards to claims 10-12, 20, 28-30 and 45-49, the Examiner argued that, in addition to elements stated for claims 1, 13 and 34, the Examiner felt that Ishige further discloses a configurable input/output characteristic that is a curvilinear compression characteristic, an input compression characteristic, and an output compression characteristic. The Examiner further argued that Ishige discloses matching the input audio signal with the narrowed dynamic range of the person fitted with the hearing aid and that, depending on the hearing loss characteristic, the Ishige device may increase or decrease the high and low frequency components at different values and that portions with lower amplitude values may be increased or decreased accordingly.

In response the Applicant respectfully submits that Ishige in no way mentions using a curvilinear compression characteristic, an input compression characteristic, and an output compression characteristic. While it is true that Ishige teaches matching the input audio signal with the narrowed dynamic range of the person fitted with the hearing aid, Ishige is totally silent on how this is done other than to apply weights to a plurality of linear phase filters, based on a frequency analysis and the user's hearing characteristics. Other than that, Ishige is totally silent on this point. Accordingly, the Applicant respectfully requests that the Examiner finds a section in Ishige that discusses these various compression characteristics or withdraws this rejection.

CLAIM REJECTIONS – 35 USC § 103

In the Office Action, the Examiner rejected claims 42 and 43 under 35 USC 103(a) as being unpatentable over Ishige (U.S. 5,892,836) in view of Martin (U.S. 6,130,950). In particular, the Examiner argued that programming the data into Ishige's memory can be done a number of ways and that Martin discloses a plurality of adjustment elements, one of which controls a gain control stage and its adjustment is stored into memory (the Examiner cited Fig. 1, col. 3 lines 65-66 and col. 4 lines 14-18 in Martin). The Examiner then argued that using a mechanical two-way switch as the adjustment elements taught by Martin would not patentably distinguish the subject claimed invention from the prior art and implementing a programming means such as the method taught by Martin or a two way switch would have been obvious to one skilled in the art.

In response, the Applicant submits that it is not possible to combine the Ishige and Martin references since this results in a modification for both devices disclosed in these references that changes the principle of operation of these devices (see sect. 2143.01 of the MPEP). Further, the Applicant submits that the devices taught in the Ishige and Martin references cannot be combined due to differences in operation of these devices.

Ishige teaches a digital hearing aid comprising a hearing compensating circuit having a transposed transversal filter, an analyzer for frequency-analyzing an input signal, a memory storing hearing characteristics of a person to be fitted with the hearing aid, and a controller receiving a frequency analysis result of the input signal and the hearing characteristics for deriving coefficients for the transposed transversal filter and supplying the derived coefficients to the transposed transversal filter. In particular, Ishige teaches the use of a plurality of linear phase filters which are weighted, based on the frequency analysis results and the hearing characteristics of the user, to come up with the transposed transversal filter.

Further, as discussed above, Ishige does not teach a user input means that the user can use to control the performance of the hearing aid during operation. The only

information obtained from the user is the user's hearing characteristics, which are stored in the memory of the digital hearing aid. This is clearly a static measurement, and does not allow the user to later provide an input signal to the hearing aid to dynamically control the performance of the hearing aid during operation.

Martin teaches a hearing aid with a microphone, a signal processing chain having a series of signal processing levels, a signal output transducer, a memory in which at least one set of parameters allocated to a hearing situation is stored that enables matching of the parameters of the signal processing stages to the hearing impairment of the wearer, and adjustment elements independent of the signal processing stage in which the allocation of the adjustment elements to the individual signal processing stages can be selectively determined. More specifically, Martin teaches a signal processing chain with a pre-amplifier, a gain control stage, a filter and a peak clipping stage. Martin also shows that the adjustment elements can be used to separately adjust the gain control stage, filter and peak clipping stages. Martin does not teach the use of several filters, or that the filter is a transposed transversal filter.

It is clear that the Martin and Ishige references cannot be readily combined since Ishige teaches combining several linear phase filters to provide a transposed transversal filter while Martin teaches the use of only one filter and does not specify that it has to be a transposed transversal filter. This is an important difference since Ishige stresses the many benefits of using a transposed transversal filter (see. lines 18-61 in Col. 4). Also, the coefficients of the transposed transversal filter taught by Ishige are weighted based on a frequency analysis of the input signal. Such a frequency analysis is not taught by Martin and not possible with the structure taught by Martin. Furthermore, the weights (which can be considered to be gains) are calculated by Ishige on a frequency basis and are related to the frequency analysis of the input signal, and the hearing characteristics of the user. Martin does not teach the calculation of any weights. Martin simply shows a gain control stage with a digital control that may be adjusted by the user. It is not clear what parameters are being controlled in the gain control stage.

Furthermore, the Applicant submits that even if one were to combine the Ishige and Martin references, the combination does not teach or suggest all claim limitations of the subject application (see sect. 2143.03 of the MPEP). In particular, claim 1 of the subject application recites a method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic. Claim 1 recites that the method comprises: (a) converting the acoustic input signal into a digital acoustic input signal; (b) transforming the digital acoustic input signal into one or more frequency domain input signals; (c) detecting the magnitude of each of the one or more frequency domain input signals; (d) receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically controlling the configuration of said input/output characteristic for loudness normalization; (e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal; (f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value; (g) transforming the one or more frequency domain output signals into a digital acoustic output signal; and (h) converting the digital acoustic output signal into the analog acoustic output signal.

The Applicant submits that it is clear that Ishige does not teach receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically controlling the configuration of said input/output characteristic for loudness normalization.

Further, the Applicant submits that Martin does not teach all of the features recited in claim 1 of the subject application. Martin does not teach transforming the digital acoustic input signal into one or more frequency domain input signals, detecting the magnitude of each of the one or more frequency domain input signals, and receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically controlling the configuration of said input/output characteristic for loudness normalization. Further, Martin does not teach determining a gain value in response to

the user adjustable digital loudness normalization control signal and the magnitude of a frequency domain input signal, or providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value, and transforming the one or more frequency domain output signals into a digital acoustic output signal.

While Martin does teach the use of adjustment elements that the user can use to control the performance of the hearing aid during operation, Martin does not teach the use of a user adjustable digital loudness normalization control signal from a user during operation for dynamically controlling the configuration of an input/output characteristic for loudness normalization. In fact, Martin does not discuss loudness normalization at all. Martin simply teaches the use of adjustment elements that can be used to separately control a gain control stage, a filter and a peak clipping stage. It is not readily apparent how this control scheme can be used by the user to control loudness normalization during operation. In fact, the Applicant submits that the average user would not know how to control gain control, filtering, or peak clipping to achieve loudness normalization. The Applicant also submits that this is a difficult thing to achieve with Martin's device since Martin does not teach a gain algorithm that can be used for loudness normalization. Further, Martin does not teach how one can use the filter to achieve loudness normalization. Also, the peak clipping stage of Martin cannot be used to control the loudness growth of an input signal, but can only be used to limit the maximum amount of the processed input signal. In addition, the user control taught by Martin appears to control static features of various signal processing components which does not lend itself to loudness normalization.

In contrast, the features recited in claim 1 of the subject application allows for a programmable input/output configuration that the user can control during operation to achieve comfortable loudness perception, and to optimally restore the loudness function to normal loudness growth (see lines 18-29 on page 13, and lines 4-12 of the subject application as filed).

The Applicant further submits that claims 13 and 33 of the subject application are independent method and apparatus claims that correspond to claim 1. The Applicant submits that since neither Ishige or Martin, alone or in combination, teach the features in claim 1 of the subject application, then these cited references cannot teach the features recited in claims 13 and 33 of the subject application.

Accordingly, based on the above discussion for the 102 and 103 rejections made by the Examiner, the Applicant respectfully submits that claims 1, 13 and 33 are novel and inventive over the cited references, and should be allowed. Furthermore, since claims 2-12, 14-32, 34-40, 42, 43 and 45-57 depend either directly or indirectly from one of claims 1, 13 and 33, and introduce other patentable features, the Applicant respectfully submits that claims 2-12, 14-32, 34-40, 42, 43 and 45-57 are also allowable.

Conclusion

In view of the foregoing comments, it is respectfully submitted that the application is now in condition for allowance. If the Examiner has any further concerns regarding the language of the claims or the applicability of the prior art, the Examiner is respectfully requested to contact the undersigned at 416-957-1603.

Respectfully submitted,

LEONARD CORNELISSE

By ______________________

Tony R. Orsi, Registration No. 55,831
Bereskin & Parr, Customer No. 001059